AVAYA

SIP Trunking

Specifications and Requirements

Invite Format / Request URI

- E.164
- 1NPANXX (For North American Dialing)

Caller ID Requirements - Outbound Calls

- E.164 (at a minimum, the country code + full number)

Supported Audio Codecs

- G.711U
- G.711A
- G.729
- G.722/G722.1

DTMF Support

- RFC2833
- SIP INFO

Note: **RFC2833** is recommended. This is the standards-based mechanism used to send DTMF digits in-band (RTP) and is supported by the majority of vendors in the industry.

SIP Support

- TCP (Port 5090)
- UDP (Port 5090)
- TLS/SRTP (Port 5091)
- SIP REFER

SIP Authentication Required for Outbound Calls

- User credentials
- Public Static IP Address from PBX / Voice Server

Supported Number of Concurrent Calls

- Unlimited

Supported Static Route - Single PBX / Voice Server

- Requires a Public Static IP address

DNS Record Types

- **A** Record
- SRV

Note: It is highly recommended to use in Inbound Calling for load balancing. This is the customer's responsibility. Lookups on the Proxy Server should be handled via Outbound Calling.

Bandwidth Requirements

- 87.2 Kbps per call leg

Quality of Service Requirements

REQUIREMENTS	DESCRIPTION
Network Delay	• To obtain toll quality, the delay cannot exceed 80 milliseconds (ms). • To obtain business communication quality, the delay must be between 80 to 180 ms. This is suitable for most enterprises. • Delays exceeding 180 ms provide a lower quality, but this may be acceptable for some enterprises.
Network Jitter	• For optimal voice quality, the average jitter must be less than half the network packet payload. This value can vary depending on the type of service the jitter buffer has in relation to other buffers, and to the packet size used. • Assuming the packet size is 20 ms, to prevent problems with voice quality, the network jitter must not exceed 20 ms.
Network Packet Loss	• To obtain toll quality, packet loss cannot exceed 1%. • For business communication quality, packet loss cannot exceed 3%. • Packet loss above 3% may result in signaling interference.

Note: The audio path for SIP Trunking goes directly from peering carriers to the customer endpoint PBX / Voice Server.